

Listing of the Claims:

The following is a complete listing of all the claims in the application, with an indication of the status of each:

- 1 1 (Previously Amended). A speech coding apparatus including at least:
 - 2 a spectrum parameter calculation section for receiving a speech
 - 3 signal, obtaining a spectrum parameter, and quantizing the spectrum
 - 4 parameter,
 - 5 an adaptive codebook section for obtaining a delay and a gain from
 - 6 a past quantized sound source signal by using an adaptive codebook, and
 - 7 obtaining a residue by predicting a speech signal, and
 - 8 a sound source quantization section for quantizing a sound source
 - 9 signal of the speech signal by using the spectrum parameter and outputting
 - 10 the sound source signal, comprising:
 - 11 a discrimination section for discriminating a voiced sound mode
 - 12 and an unvoiced sound mode on a basis of a past quantized gain of an
 - 13 adaptive codebook;
 - 14 a sound source quantization section which has a codebook for
 - 15 representing a sound source signal by a combination of a plurality of non-
 - 16 zero pulses and collectively quantizing amplitudes or polarities of the
 - 17 pulses based on an output from said discrimination section, and searches
 - 18 combinations of code vectors stored in said codebook and a plurality of
 - 19 shift amounts used to shift positions of the pulses so as to output a
 - 20 combination of a code vector and shift amount which minimizes distortion
 - 21 relative to input speech; and
 - 22 a multiplexer section for outputting a combination of an output
 - 23 from said spectrum parameter calculation section, an output from said
 - 24 adaptive codebook section, and an output from said sound source
 - 25 quantization section.

1 2 (Previously Amended). A speech coding apparatus including at least:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter,
4 an adaptive codebook section for obtaining a delay and a gain from
5 a past quantized sound source signal by using an adaptive codebook, and
6 obtaining a residue by predicting a speech signal, and
7 a sound source quantization section for quantizing a sound source
8 signal of the speech signal by using the spectrum parameter and outputting
9 the sound source signal, comprising:
10 a discrimination section for discriminating a voice sound mode and
11 an unvoiced sound mode on a basis of a past quantized gain of an adaptive
12 codebook;
13 a sound source quantization section which has a codebook for
14 representing a sound source signal by a combination of a plurality of non-
15 zero pulses and collectively quantizing amplitudes or polarities of the
16 pulses based on an output from said discrimination section, and outputs a
17 code vector that minimizes distortion relative to input speech by generating
18 positions of the pulses according to a predetermined rule; and
19 a multiplexer section for outputting a combination of an output
20 from said spectrum parameter calculation section, an output from said
21 adaptive codebook section, and an output from said sound source
22 quantization section.

1 3 (Previously Amended). A speech coding apparatus including at least:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter,
5 an adaptive codebook section for obtaining a delay and a gain from a
6 past quantized sound source signal by using an adaptive codebook, and
7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source
9 signal of the speech signal by using the spectrum parameter and outputting
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and
12 an unvoiced sound mode on the basis of a past quantized gain of an
13 adaptive codebook;

14 a sound source quantization section which has a codebook for
15 representing a sound source signal by a combination of a plurality of non-
16 zero pulses and collectively quantizing amplitudes or polarities of the
17 pulses based an output from said discrimination section, and a gain
18 codebook for quantizing gains, and searches combinations of code vectors
19 stored in said codebook, a plurality of shift amounts used to shift positions
20 of the pulses, and gain code vectors stored in said gain codebook so as to
21 output a combination of a code vector, shift amount, and gain code vector
22 which minimizes distortion relative to input speech; and

23 a multiplexer section for outputting a combination of an output
24 from said spectrum parameter calculation section, an output from said
25 adaptive codebook section, and an output from said sound source
26 quantization section.

1 4 (Previously Amended). A speech coding apparatus including at least:

2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter,

5 an adaptive codebook section for obtaining a delay an a gain from a
6 past quantized sound source signal by using an adaptive codebook, and
7 obtaining a residue by predicting a speech signal, and

8 a sound source quantization section for quantizing a sound source
9 signal of the speech signal by using the spectrum parameter and outputting
10 the sound source signal, comprising:

11 a discrimination section for discriminating a voice sound mode and
12 an unvoiced sound mode on the basis of a past quantized gain of an
13 adaptive codebook;
14 a sound source quantization section which has a codebook for
15 representing a sound source signal by a combination of a plurality of non-
16 zero pulses and collectively quantizing amplitudes or polarities of the
17 pulses based on an output from said discrimination section indicates a
18 predetermined mode, and a gain codebook for quantizing gains, and
19 outputs a combination of a code vector and gain code vector which
20 minimizes distortion relative to input speech by generating positions of the
21 pulses according to a predetermined rule; and
22 a multiplexer section for outputting a combination of an output
23 from said spectrum parameter calculation section, an output from said
24 adaptive codebook section, and an output from said sound source
25 quantization section.

5 (Canceled).

1 6 (Previously Amended). A speech coding/decoding apparatus comprising:
2 a speech coding apparatus including:
3 a spectrum parameter calculation section for receiving a speech
4 signal, obtaining a spectrum parameter, and quantizing the spectrum
5 parameter,
6 an adaptive codebook section for obtaining a delay and a gain from
7 a past quantized sound source signal by using an adaptive codebook, and
8 obtaining a residue by predicting a speech signal,
9 a sound source quantization section for quantizing a sound source
10 signal of the speech signal by using the spectrum parameter and outputting
11 the sound source signal,
12 a discrimination section for discriminating a voice sound mode and

13 an unvoiced sound mode on the basis of a past quantized gain of a adaptive
14 codebook, and
15 a codebook for representing a sound source signal by a
16 combination of a plurality of non-zero pulses and collectively quantizing
17 amplitudes or polarities of the pulses when an output from said
18 discrimination section indicates a predetermined mode,
19 said sound source quantization section searching combinations of
20 code vectors stored in said codebook and a plurality of shift amounts used
21 to shift positions of the pulses so as to output a combination of a code
22 vector and shift amount which minimizes distortion relative to input
23 speech, and further including
24 a multiplexer section for outputting a combination of an output
25 from said spectrum parameter calculation section, an output from said
26 adaptive codebook section, and an output from said sound source
27 quantization section; and
28 a speech decoding apparatus including at least:
29 a demultiplexer section for receiving and demultiplexing a
30 spectrum parameter, a delay of an adaptive codebook, a quantized gain,
31 and quantized sound source information,
32 a mode discrimination section for discriminating a mode by using a
33 past quantized gain in said adaptive codebook,
34 a sound source signal reconstructing section for reconstructing a
35 sound source signal by generating non-zero pulses from the quantized
36 sound source information when an output from said discrimination
37 indicates a predetermined mode, and
38 a synthesis filter section which is constituted by spectrum
39 parameters and reproduces a speech signal by filtering the sound source
40 signal.

1 7 (Previously Amended). A speech coding/decoding apparatus comprising:
2 a speech coding apparatus including:
3 a spectrum parameter calculation section for receiving a speech
4 signal, obtaining a spectrum parameter, and quantizing the spectrum
5 parameter,
6 an adaptive codebook section for obtaining a delay and a gain from
7 a past quantized sound source signal by using an adaptive codebook, and
8 obtaining a residue by predicting a speech signal,
9 a sound source quantization section for quantizing a sound source
10 signal of the speech signal by using the spectrum parameter and outputting
11 the sound source signal,
12 a discrimination section for discriminating a voice sound mode and
13 an unvoiced sound mode on the basis of a past quantized gain of an
14 adaptive codebook, and
15 a codebook for representing a sound source signal by a
16 combination of a plurality of non-zero pulses and collectively quantizing
17 amplitudes or polarities of the pulses based on an output from said
18 discrimination section,
19 said sound source quantization section outputting a combination of
20 a code vector and shift amount which minimizes distortion relative to input
21 speech by generating positions of the pulses according to a predetermined
22 rule, and further including
23 a multiplexer section for outputting a combination of an output
24 from said spectrum parameter calculation section, an output from said
25 adaptive codebook section, and an output from said sound source
26 quantization section; and
27 a speech decoding apparatus including at least:
28 a demultiplexer section for receiving and demultiplexing a
29 spectrum parameter, a delay of an adaptive codebook, a quantized gain,
30 and quantized sound source information,

31 a mode discrimination section for discriminating a mode by using a
32 past quantized gain in said adaptive codebook,
33 a sound source signal reconstructing section for reconstructing a
34 sound source signal by generating positions of pulses according to a
35 predetermined rule and generating amplitudes or polarities for the pulses
36 from a code vector when an output from said discrimination section
37 indicates a predetermined mode, and
38 a synthesis filter section which includes spectrum parameters and
39 reproduces a speech signal by filtering the sound source signal.

1 8 (Previously Amended). A speech coding apparatus comprising:
2 a spectrum parameter calculation section for receiving a speech
3 signal, obtaining a spectrum parameter, and quantizing the spectrum
4 parameter;
5 means for obtaining a delay and a gain from a past quantized sound
6 source signal by using an adaptive codebook, and obtaining a residue by
7 predicting a speech signal; and
8 mode discrimination means for receiving a past quantized adaptive
9 codebook gain and performing mode discrimination associated with a
10 voiced/unvoiced mode by comparing the gain with a predetermined
11 threshold, and
12 further comprising:
13 sound source quantization means for quantizing a sound source
14 signal of the speech signal by using the spectrum parameter and outputting
15 the signal, and searching combinations of code vectors stored in a
16 codebook for collectively quantizing amplitudes or polarities of a plurality
17 of pulses in a predetermined mode and a plurality of shift amounts used to
18 temporally shift a predetermined pulse position so as to select a
19 combination of an index of a code vector and a shift amount which
20 minimizes distortion relative to input speech;

21 gain quantization means for quantizing a gain by using a gain
22 codebook; and
23 multiplex means for outputting a combination of outputs from said
24 spectrum parameter calculation means, said adaptive codebook means, said
25 sound source quantization means, and said gain quantization means.

1 9 (Original). An apparatus according to claim 8, wherein said sound source
2 quantization means uses a position generated according to a predetermined
3 rule as a pulse position when mode discrimination indicates a
4 predetermined mode.

1 10 (Original). An apparatus according to claim 9, wherein when mode
2 discrimination indicates a predetermined mode, a predetermined number of
3 pulse positions are generated by random number generating means and
4 output to said sound source quantization means.

1 11 (Original). An apparatus according to claim 8, wherein when mode
2 discrimination indicates a predetermined mode, said sound source
3 quantization means selects a plurality of combinations from combinations
4 of all code vectors in said codebook and shift amounts for pulse positions
5 in an order in which a predetermined distortion amount is minimized, and
6 outputs the combinations to said gain quantization means, and
7 said gain quantization means quantized a plurality of sets of
8 outputs from said sound source quantization means by using said gain
9 codebook, and selects a combination of a shift amount, sound source code
10 vector, and gain code vector which minimizes the predetermined distortion
11 amount.